or made public. Thus, Figure 2 is <u>not</u> a prior art as far as the instant application is concerned. Accordingly, Figure 2 should <u>not</u> be marked as "Prior Art" as suggested by the Examiner.

At section 3, the Examiner states that the full name of each inventor has not been set forth.

It is respectfully submitted, the two inventors of the present invention are *Ye Wang* (*Ye* is the first name and *Wang* is the last name) and *Miikka Vilermo* (*Miika* is the first name and *Vilermo* is the last name), as shown in the updated filing receipt mailed September 17, 2002.

The Examiner also states that the oath or declaration is not signed by *Mauri Vaananen* and *Leonid Yaroslavsky*.

It is respectfully submitted that although the names of *Mauri Vaananen* and *Leonid Yaroslavsky* appear on the cover sheet of the patent application, *Mauri Vaananen* and *Leonid Yaroslavsky* are not named inventors. The names of *Mauri Vaananen* and *Leonid Yaroslavsky* are not listed in the as-filed Declaration and Power of Attorney, a copy of which is attached herein.

At section 4, claims 1-8, 10 and 12-17 are rejected under 35 U.S.C. 103(a) as being unpatentable over *Miyamori et al.* (U.S. Patent No. 5,737,720, hereafter referred to as *Miyamori*) in view of *Chen et al* ("Video Compression Using Integer DCT" Image Processing, 2000, Proceeding 2000 International Conference, vol.2, pp. 844-845.)

In rejecting claims 1 and 10, the Examiner states that *Miyamori* teaches low bit rate multichannel audio coding methods using non-linear adaptive bit allocation. Specifically, Miyamori teaches coding audio signals in a sound system having a plurality of sound channels for providing M sets of audio signals, wherein a plurality of intra-channel signal redundancy removal devices are used to reduce the audio signal for providing first signals indicative of the reduced audio signals (col.7, line 8 - col.9, line 60).

It is respectfully submitted that *Miyamori* discloses a low-bit rate encoder for compression-encoding digital audio signals of a plurality of channels making use of both the property of the audio signal and the hearing sense of the human being. To that end the encoder

includes an energy detecting section for detecting energies of the digital audio signals in the channels, and a bit allocation module for determining bit allocation amounts to the respective channels based on the detected results. As such, the relationship between the energy and the bit allocation amount represents a non-linear characteristic so as to eliminate the redundancy of bit allocation amount in the compression encoding of the multi-channel system. *See Abstract*. Col. 7, line 8 to col. 8, line 40 of *Miyamori* describes the encoder shown in Figure 1. In particular, the encoder comprises a plurality of amplitude information detection circuits 200, operatively connected to the plurality of channels, for detecting the energies of the digital audio signals, or the peak values of amplitude information from quantized signals of respective channels for every period corresponding to the number of samples of audio data processed at a time (col.7, lines 19-32). Based on the detection results, bit allocating determining circuits 500 allocate the bit amounts to respective encoding elements 400 for every respective channel from peak values of every respective channel(col.7, line 33-38). As such, intra-channel redundancy may be removed. The delay lines 300 are used to delay the arrival at the encoding elements 400 of the audio signals from the sampling & quantization elements 100, so that the allocated bit amounts conveyed to the encoding elements match the period in which the energies are detected by the amplitude information detection circuits. No inter-channel redundancy removal is disclosed or even suggested here.

Col. 8, line 41 - col. 9, line 44 describes an ATRAC system, which makes use of the hearing sense characteristics of the human being. The ATRAC system is implemented in the encoding elements 400 for compression-encoding using a plurality of MDCT circuits. Although this compression-encoding technique is useful in compressing audio signals of stereo 2 channels at a fixe bit rate by making use of the hearing sense characteristics, the described technique and encoder does not suggest removal of inter-channel signal redundancy.

Accordingly, the Examiner admits that *Miyamori* does not specifically teach implementation of reducing the inter-channel signal redundancy in the second signals, which are indicative of audio data of integers. The Examiner further states that data reduction of integers is well known in the art. However, the Examiner fails to address the issue of inter-channel signal redundancy.

The Examiner cites *Chen* for teaching integer discrete cosine transform (IntDCT) in an MPEG coder. It is respectfully submitted that IntDCT is used in the claimed invention to

remove intra-channel signal redundancy. However, *Chen* does not disclose or even suggest inter-channel signal redundancy removal.

For the above reasons, claims 1 and 10 are distinguishable over the cited *Miyamori* and *Chen* references.

As for claims 2-8 and 12-17, they are dependent from claims 1 and 11 and recite features not recited in claims 1 and 10. For reasons regarding claims 1 and 11 above, it is respectfully submitted that claims 2-8 and 12-17 are also distinguishable over the cited *Miyamori* and *Chen* references.

CONCLUSION

Claims 1-8, 10 and 12-17 are distinguishable over the cited *Miyamori* and *Chen* references. Claims 9 and 11 have been rewritten in independent form and in condition for allowance. Early allowance of claims 1-17 is earnestly solicited.

Respectfully submitted,

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9. (Amended) A method of coding audio signals in a sound system having a plurality of sound channels for providing *M* sets of audio signals from input signals, wherein *M* is a positive integer greater than 2, and wherein a plurality of intra-channel signal redundancy removal devices are used to reduce the audio signals for providing first signals indicative of the reduced audio signals, said method comprising the steps of:

converting the first signals to audio data of integers for providing second signals indicative of the audio data; and

reducing inter-channel signal redundancy in the second signals for providing third signals indicative of the reduced second signals[The method of claim 1], wherein the second signals are divided into a plurality of scale factor bands and the third signals are divided into a plurality of corresponding scale factor bands, said method further comprising the step of comparing coding efficiency in the second signals to coding efficiency in the third signals in corresponding scale factor bands, for bypassing the reducing step if the coding efficiency in the third signals is smaller than the coding efficiency in the second signals.

11. (Amended) A system for coding audio signals in a sound system having a plurality of sound channels for providing *M* sets of audio signals from input signals, wherein *M* is a positive integer greater than 2, and wherein a plurality of intra-channel signal redundancy removal devices are used to reduce the audio signals for providing first signals indicative of the reduced audio signals, said system comprising:

a first means, responsive to the first signals, for converting the first signals to audio data of integers for providing second signals indicative of the audio data; and

a second means, responsive to the second signals, for reducing inter-channel signal redundancy in the second signals for providing third signals indicative of the reduced second signals[The system of claim 10], wherein the second signals are divided into a plurality of scale factor bands and the third signals are divided into a plurality of corresponding scale factor bands, and wherein coding efficiency in the second signals in a scale factor band is representable by a first value and coding efficiency in the third signals in the corresponding scale factor band is representable by a second value, said system further comprising a comparison means, responsive to the second and third signals, for bypassing the inter-channel signal redundancy

reduction in said scale band factor by the second means when the first value is greater or equal to the second value.